

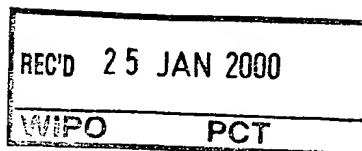


The
Patent
Office



INVESTOR IN PEOPLE

The Patent Office
Concept House
Cardiff Road
Newport
South Wales
NP10 8QQ



GB 99/3753

EU

I, the undersigned, being an officer duly authorised in accordance with Section 74(1) and (4) of the Deregulation & Contracting Out Act 1994, to sign and issue certificates on behalf of the Comptroller-General, hereby certify that annexed hereto is a true copy of the documents as originally filed in connection with the patent application identified therein.

In accordance with the Patents (Companies Re-registration) Rules 1982, if a company named in this certificate and any accompanying documents has re-registered under the Companies Act 1980 with the same name as that with which it was registered immediately before re-registration save for the substitution as, or inclusion as, the last part of the name of the words "public limited company" or their equivalents in Welsh, references to the name of the company in this certificate and any accompanying documents shall be treated as references to the name with which it is so re-registered.

In accordance with the rules, the words "public limited company" may be replaced by p.l.c., plc, P.L.C. or PLC.

Re-registration under the Companies Act does not constitute a new legal entity but merely subjects the company to certain additional company law rules.

**PRIORITY
DOCUMENT**

SUBMITTED OR TRANSMITTED IN
COMPLIANCE WITH RULE 17.1(a) OR (b)

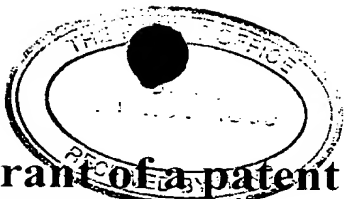
Signed

Dated

14.1.2000

**CERTIFIED COPY OF
PRIORITY DOCUMENT**

This Page Blank (uspto)



Request for grant of a patent

(See the notes on the back of this form. You can also get an explanatory leaflet from the Patent Office to help you fill in this form.)

12NGU98 E404166-1 D01631
P01/7700 3.00 - 9824776.0

The Patent Office

Cardiff Road
Newport
Gwent NP9 1RH

Fee: £0

1. Your reference

AJR/40273

2. Patent application number
(The Patent Office will fill in this part)

9824776.0

11 NOV 1998

3. Full name, address and postcode of the or of each applicant (underline all surnames)

KEMP, Michael Joseph
a British citizen
of Casa Lucia
Vale Formosilho
S Marcos da Serra
8375 S B de Messines
PORTUGAL

Patents ADP number (if you know it)

If the applicant is a corporate body, give the country/state of incorporation

7046036001

4. Title of the invention

AUDIO DYNAMIC CONTROL EFFECTS SYNTHESISER

5. Name of your agent (if you have one)

Reddie & Grose
16 Theobalds Road
LONDON
WC1X 8PL

"Address for service" in the United Kingdom to which all correspondence should be sent (Including the postcode)

Patents ADP number (if you know it)

91001

6. If you are declaring priority from one or more earlier patent applications, give the country and the date of filing of the or of each of these earlier applications and (if you know it) the or each application number

Country

Priority application
(If you know it)

Date of filing
(day/month/year)

7. If this application is divided or otherwise derived from an earlier UK application, give the number and the filing date of the earlier application

Number of earlier application

Date of filing
(day/month/year)

8. Is a statement of inventorship and of right to grant of a patent required in support of this request? (Answer 'Yes' if:

a) any applicant named in part 3 is not an inventor, or

b) there is an inventor who is not named as an applicant, or

c) any named applicant is a corporate body.

See note (d))

YES

Patents Form 1/77

9. Enter the number of sheets of any of the following items you are filing with this form. Do not count copies of the same document.

Continuation sheets of this form

Description	24
Claim(s)	
Abstract	
Drawing(s)	13

10. If you are also filing any of the following, state how many against each item.

Priority documents

Translations of priority documents

Statement of inventorship and right to grant of a patent (*Patents Form 7/77*)

Request for preliminary examination and search (*Patents Form 9/77*)

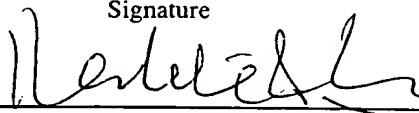
Request for substantive examination (*Patents Form 10/77*)

Any other documents
(please specify)

11. I/We request the grant of a patent on the basis of this application.

Signature

Date



11 November 1998

12. Name and daytime telephone number of person to contact in the United Kingdom
- A J ROBSON
0171-242 0901

Warning

After an application for a patent has been filed, the Comptroller of the Patent Office will consider whether publication or communication of the invention should be prohibited or restricted under Section 22 of the Patents Act 1977. You will be informed if it is necessary to prohibit or restrict your invention in this way. Furthermore, if you live in the United Kingdom, Section 23 of the Patents Act 1977 stops you from applying for a patent abroad without first getting written permission from the Patent Office unless an application has been filed at least 6 weeks beforehand in the United Kingdom for a patent for the same invention and either no direction prohibiting publication or communication has been given, or such direction has been revoked.

Notes

- If you need help to fill in this form or you have any questions, please contact the Patent Office on 0645 500505.
- Write your answers in capital letters using black ink or you may type them.
- If there is not enough space for all the relevant details on any part of this form, please continue on a separate sheet of paper and write "see continuation sheet" in the relevant part(s). Any continuation sheet should be attached to this form.
- If you have answered 'Yes' Patents Form 7/77 will need to be filed.
- Once you have filled in the form you must remember to sign and date it.
- For details of the fee and ways to pay please contact the Patent Office.

Audio Dynamic Control Effects Synthesiser

Patent Application PCT GB97/02159 (WO98/0714) ("The Prior Application") describes an audio effects synthesiser with or without analyser and should be read
5 along with this description of further improvements.

The purpose of this invention is to store the characteristics of audio level control devices which have audibly desirable properties and to be able to synthesise these properties at will.

Summary of Figures

Figure 1 shows an analysis arrangement for a device under test (D.U.T.).

Figure 2 illustrates two characteristics of a compressor device namely the gain
15 characteristic and the time dependent characteristic.

Figure 3 shows a flow diagram of the process of analysing the gain characteristic of a D.U.T.

Figure 4 shows a table of derived figures from a hypothetical 'ideal' compressor and some example figures which may be obtained in reality.

20 Figure 5 shows graphically the values of figure 4.

Figure 6 illustrates the derivation of intermediate ratio curves from measured curves.

Figure 7 shows the application of test impulses as required by The Prior Application in conjunction with a gain varying device.

Figure 8 shows a flow diagram for assessing the time constant characteristic of a
25 D.U.T.

Figure 9 shows an arrangement for applying the simulation of a gain varying device in conjunction with the simulation of non-linearity described in The Prior Application.

Figure 10 shows a flow diagram of the implementation of the time constant characteristics of a device during simulation.

30 Figure 11 shows an example of an alternative way of collecting a set of impulse response data to simulate novel compression effects.

Figure 12 shows another example of an alternative way of collecting sets of impulse response data to further simulate compression effects.

Figure 13 shows in more detail the bi-linear interpolation method.

5 Characteristics of a Dynamic Range controlling Device.

Figure 1 shows a typical arrangement of an audio compressor device 1 (in this case the device under test), being driven by a signal generating arrangement 2, and being analysed by the signal analysis arrangement 3.

10

A compressor is designed to reduce the dynamic range of audio program material by reducing the amplitude of louder passages in relation to that of quieter passages. In a typical compressor as shown at 1, the input signal is fed (often with some pre-amplification and or variable attenuation) to a gain control element 11. This alters the
15 amplitude of the signal which is then fed again usually via some amplification or buffering to the output where there is usually some additional variable gain available to compensate for attenuation in the overall signal path.

20

Some of the output signal is usually fed to a rectification circuit where controls 12 and 13 select a threshold above which gain reduction is to be effected and an amount or ratio by which gain is to be reduced according to the output signal. The rectified signal is further applied to a time constant arrangement where controls 14 and 15
25 apply attack and recovery time constants by which gain is reduced and restored when the output signal increases or decreases.

25

Note that there are a number of variations possible in this arrangement and also it is possible to rearrange the gain control circuitry so that dynamic range expansion is produced or so that signal gating may occur. These do not affect the operation of the analysis device.

30

The operation of the device under test is determined by the signal generating arrangement 2 comprising the choice of a continuous signal generator 21 or a pulse

generator 22 fed to a variable gain control element 23 and then to the device under test via a digital to analogue converter 24.

5 Note that the D-A converter may be omitted if the device under test is actually a digital device or if the signal generating arrangement is implemented in analogue circuitry. For the rest of this discussion the assumption is that the device under test is analogue and the analysis and synthesis device is digital.

10 The output of the device under test is fed to the signal analysis arrangement 3 comprising an analogue to digital converter 31 and the choice of level detection circuitry 32 and impulse storage circuitry 33.

15 The entire analysis arrangement is controlled by system controller 34 which is typically a data processing arrangement which may share processing hardware with various elements of the complete analysis and synthesis system.

Figure 2 shows the main characteristics of the device under test to be assessed.

20 Fig 2 (a) shows the gain characteristic of a typical compressor. Curve 41 shows the relationship between the input level of a signal fed to the device and the resulting output at a particular setting of the controls. It can be seen that at low levels the output rises linearly in proportion to the input, along the dotted line 43 such that a 1dB increase in input results in a 1dB increase in output.

25 As the input level approaches the threshold setting 42 the gain begins to modify to approach that of line 44 which represents the ratio or slope of the compressor. This may be for example that it requires a 3 dB increase in input to produce a 1 dB increase in output (a ratio of 3:1). There is usually no sudden transition at the threshold and the degree of 'softness' of the knee of the curve is one distinguishing
30 characteristic of a particular compressor. It is also possible to have multiple thresholds between multiple ratios.

Fig 2 (b) shows the time constant characteristic of a typical compressor. The input signal is shown as a continuous sine wave which doubles in amplitude (increases by 6dB) at 45 and then decreases by 6dB at 46. The corresponding output shows that there is an initial increase by 6dB at 47 following which the gain is reduced according to the attack time constant until it reaches an essentially stable reduced gain at 49 determined from the curve of part (a). When the input signal reduces by 6dB the output also reduces immediately by 6dB at 48 then the gain is increased according to the release time constant until the appropriate gain for the input as determined by the curve in part (a) is restored by point 50.

Time constants are typically operator variable and the range and options provided by a particular device contribute to the user perceived operability of the device and to the audible characteristics of the device under test. However it is not essential to measure these characteristics as a standard set of time constants can be provided which are common to most devices to be simulated. In the event that it is desired to assess this characteristic of a device under test to more closely simulate a specific device the method is described below.

The third characteristic that determines the audible effect of a gain control device is the distortion or non-linearity caused by the gain control element. This may be assessed using the means of The Prior Application with some small modifications as set out below.

Analysis of the gain characteristic.

To properly analyse a device under test to determine its gain characteristic it is necessary to take at least one set of measurements corresponding to a specific setting of the ratio control on the device. As is described later the characteristic of other ratios may be interpolated between any pair of measured ratio characteristics and it is also possible to further interpolate between a measured ratio characteristic and an ideal 1:1 (no compression) and infinity:1 (limiting compression) characteristic.

In this example three sets of measurements are taken, for example for a ratios of 1.5:1, 5:1 and 20:1 (although any value or values spanning commonly used ratios may be chosen). To take each set of measurements the attack and decay settings of the device should each be set to either their maximum settings or to 1 second, whichever is shorter.

To take the set of measurements corresponding to the ratio of 1.5:1, this setting is selected on the device under test, and if a threshold control is available this is set to -20dB, and the signal generator of figure 1 is set to generate a 1kHz sine wave tone (although other frequencies may be used as described later) at a typical operating level for the device, chosen to be suitably below the overload limit of the device. For a professional unit a level of 0dBm is appropriate. If available any make-up or output gain of the device under test should now be adjusted so that the level returned to the level detector of figure 1 is also at 0dBm once the gain of the device under test has reached a steady state. If such an output adjustment is not available and this level is not achievable it is permissible to use any appropriate output level from the device.

Figure 3 shows a flow diagram of the test procedure which the signal generator and level detector conduct under computer control once the above conditions are set up. A sequence of measurements are taken for signals fed to the device under test between -40dBm and 0dBm at 1 dB intervals. As each level generated is set up, the output level from the device under test is repeatedly assessed until this level becomes stable to within 0.1dB over a 100mS interval. The output level is then recorded for this input level.

The amplitude interval with which the tests are conducted is dependent on the accuracy required in the simulations. The 1dB steps described give good results for most applications but if there are time constraints on taking the tests optionally larger steps may be used and data may be interpolated to make up for missing detail.

The set of device output levels corresponding to each input levels step is recorded within the memory of the computer and may be represented by the first column of table (a) of figure 4.

- 5 The above set of tests may be repeated for as many ratio settings as desired, for example for ratios of 5:1 and 20:1, shown in the second and third columns of table (a) of figure 4.

10 The number of different sets of measurements corresponding to each ratio setting to be taken depend on the degree of accuracy to which it is required to simulate the device under test and also the number of options available on the device. Simple devices may only have one or two slope options in which case these provide sufficient data for storing the characteristics of this device. The later description of the use of this data shows how available data is interpolated to provide options sometimes
15 beyond what was available with the device under test.

Figure 4 (a) shows an example table of data derived from the device under test with some entries omitted for clarity. In fact these figures would be derived from a device with a 'perfect' gain characteristic as can be seen from the plot of these figures in
20 figure 5 (a) which shows that there is no 'soft knee' as would be encountered in reality. Such a device may not sound ideal but serves to illustrate the process more clearly. It should also be clear that the ratio setting of the device means the change in output level divided by change in input level expressed in dB provided that the signal is well above the threshold.

25

(ratio = (delta dB in) / (delta dB out)).

If the ratio settings are not specified on the device under test the analyser may assess the curve by inspection of the data at the higher levels applied and assign a value to
30 the ratio based on the above formula. It is also possible to store the actual text used to describe the ratios on the device under test (provided the operator enters this data) and

provide these options during simulation rather than the derived figures as an optional closer match to the user interface of the device under test.

These figures also show an exact setting of threshold at -20dBm at the input to the device. In practice if this varies with ratio settings the figures will not be so clear to interpret but the following processing still applies and the variations in the device will remain embedded within the data for simulation. If a calibrated threshold setting is provided but -20dB is not available, it is acceptable to use another convenient threshold setting and ideally the analyser will be arranged to allow this value to be stored along with the analysis data. This threshold figure may then be used during simulation along with the operator's desired threshold to achieve an accurate simulation by adjustment of the values as described below.

Figure 4 (b) shows some more typical data which may be obtained from a real device and this serves to illustrate the next stage of processing.

Although figures are obtained for an input range of 40dB these figures are extrapolated by determining the actual slope at the upper end of the graph by averaging over the 3 or 4 highest measurements and using this slope to generate a notional straight line extrapolation at any further higher input level. At the lower end of the graph a slope of unity is assumed and this is used to extrapolate down to arbitrary low levels of input.

Simulation of the gain characteristic

During simulation of the gain characteristic of the device which was under test, the operator must first select a desired ratio R of compression to be applied to the signal to be processed by the simulation.

As the input signal is processed the amplitude is continually assessed according to details to be described. The input level is thus determined to be say I dB and this is indicated in figure 5(b).

If the value R matches the ratio of one of the stored curves, say 5:1, the desired output level may be immediately assessed by inspection of the data, and if the value I falls between two stored input values (say I1 and I2 in the diagram) a simple linear
5 interpolation may be used to determine the desired output P. From the known input and desired output an attenuation G (which may be negative, i.e. an attenuation) may be simply calculated as

$$G = P - I,$$

10 and applied to the input signal as described below.

If the value R does not match the ratio of one of the stored curves but there is data for ratios greater and less than the value R then the desired output level can be
15 determined by interpolation between the two stored ratios as follows with reference to figure 6.

Figure 6 shows a graphical representation of typical stored data where the curve for 5:1 is known (curve 73) and also that for 20:1 (curve 72). It is desired to interpolate
20 curve 71 representing a ratio of 10:1 from this data. Also shown are ideal curves 1:1 (straight line 75) and infinity:1 (curve 74 which has a sharp knee where it meets curve 75 and progresses down at unity gain ratio along line 75).

The curves shown incorporate various approximations inherent in the device
25 originally tested to produce this data including errors in ratio, differences in threshold and small offsets in gain at low level as would be expected from real devices. The interpolation described reflects these imperfections and provides a smooth transition from one curve to another under user control. Any interpolation towards one of the ideal curves will again produce a smooth transition from the measured curve to the
30 ideal curve.

The following describes in detail the interpolation steps.

If curves are known for the higher ratio of M:1 and the lower ratio of N:1, and we desire to know the required output level at a given input level with a ratio of R:1, first determine the coefficient of linear interpolation a from the formula

5

$$a = N(M-R) / (R(M-N)). \quad (1)$$

Then if the output level at the higher ratio M:1 would be X dBm, and the output level at the lower ratio would be Y dBm determined from the relevant curves (using linear interpolation if necessary as described above), the output level Z we require for ratio R is

10

$$Z = (1-a)X + aY. \quad (2)$$

15 Therefore, first considering the example based on ideal curves where M=20 and N=5 are known curves (72 and 73) and it is desired to determine the output for a curve of R=10 (curve 71), substituting in equation 1, a is determined to be as follows:

$$a = N(M-R) / (R(M-N)) = 5(20-10) / (10(20-5)) = 5 \times 10 / (10 \times 15) = 1/3.$$

20

Referring again to figure 6 the dotted curve 71 representing a ratio of 10:1 may thus be derived by using the $a = 1/3$ at any input level to perform a linear interpolation between the two curves 72 and 73 according to formula (2) where X is the output level from curve 72 and Y is the output level from curve 73.

25

Note that although this interpolated curve may not be identical to a measured curve for this ratio from a device under test, the interpolated value satisfies the two criteria that (i) it does represent a reasonable derivation of the unmeasured value and as R is varied between the ratios of the two curves the interpolated curve smoothly transforms from one of the extreme curves to the other, and (ii) in the case of idealised curves the method produces an exact solution.

30

If the value R does not lie between two stored ratios one approach is to deny the setting to the user, offering the nearest alternative stored ratio as it was not possible to determine what the device under test would have provided for ratio R. Alternatively a desired ratio may be derived by interpolation using the above method between the
5 nearest known ratio and (for a larger ratio) the ideal curve for an infinite ratio, as shown at 74 in figure 6, and (for a smaller ratio) a unity gain curve representing a 1:1 ratio, as shown at 75 in figure 6. In the case of interpolating to an infinite ratio, the formula (1) needs to be adjusted as follows:

10 $a=N/R.$ (3)

Thus a is unity if R is chosen to equal N as expected and a falls to zero as R tends towards infinity.

15 Where interpolation is towards a 1:1 ratio, equation 1 also simplifies with N=1 to

$$a=(M-R)/(R(M-1)). \quad (4)$$

Thus when R is equal to M, a is zero, so the curve tends to that of M, and when R is 1,
20 a is 1, thus the curve becomes exactly the 1:1 curve and no compression effect results.

The interpolation formulae (2) may also be applied directly to gains derived from each ratio rather than absolute output level for any given input level as the formula is independent of any constant added to each term, X, Y and Z.

25

Determination of Time Dependent Characteristics

In order to determine the time constant characteristic of the device under test it must be analysed for dynamic response. It is possible to take a number of test results from
30 the device at different ratio settings if it is desired to simulate the device at greatest accuracy but generally the same time constants apply at all ratios and so it is possible to make the measurements at one fairly high ratio, for example 20:1.

The device under test should thus be set to a 20:1 ratio with a threshold of -20dBm or as close to this as possible and the signal generator 2 of figure 1 is set to produce a 0dBm tone at typically 1kHz. The output or make-up gain should be adjusted for a -
5 20dB output from the device under test or as close as practical so that overshoots during test do not cause distortion.

The controls are first set to the fastest attack and release times available on the device under test, and then the analysis process is started and proceeds automatically as
10 indicated in figure 8.

It is assumed that the output of the test generator is already at 0dBm into the device under test. The test output is first reduced to -20dBm (step 80) and the output from the device is repeatedly assessed (step 81) in the level detector 3 of figure 1 until the
15 level has stabilised to a variation of less than 0.1dB over the period of typically 1 second. The variation in output level from the device is stored repeatedly during this process to determine the dynamic characteristic of the recovery time constant of the device.

20 The generator output is now increased again to 0dBm (step 82) and again the output of the device is repeatedly assessed and stored (step 83) until the output is again stable to within 0.1dB per second.

A number of ways of determining level during a rapidly changing signal are possible
25 but since the signal generator and level detector are under full control of the system controller one method with fast response is to modify the frequency of the test tone (if necessary) so that it is synchronous with the sampling rate of the generator/detector combination. It can thus be arranged that in the detector circuitry the maximum level reached in each cycle is dependent solely on the total gain of the device and this value
30 can be recorded for each cycle of the test signal recovered for the device as an indicator of amplitude of the signal.

From these measurements the time constant characteristic for the fast setting of the device under test can be determined and stored. All the gain values for this derivation are considered as multiplicative factors, not as logarithmic gain changes. The time constant is defined as follows: During a change in gain from G_1 to G_2 it is the time taken for the gain to reach G_3 , where $(G_3 - G_2) / (G_1 - G_2) = 1/e$, where e is the exponential constant. Since the initial and final gain is known the period to reach this gain may be calculated from measured characteristic. This derivation applies when the input signal is increased suddenly in level. When the input signal is reduced suddenly, the gain recovery is usually exponential towards the gain setting applicable for zero input signal, so G_2 should be taken as this limiting gain and if the gain does not lie between G_1 and G_2 under these conditions the time constant should be derived from a smaller change in gain according to the assumption that the gain follows a similar exponential curve.

It is also possible that a multi-stage time constant circuit is included, and inspection of the amplitude change characteristic can determine this for use in simulation.

Many existing devices offer a limited set of attack and recovery time constants, often labelled as fast/medium/slow and thus the measurements above should be repeated for each of these settings. In simulation the operator can be offered these simple choices which can recall the stored time constants when selected. In addition the user can be offered a continuously variable time constant to extend the simulation beyond that achievable with the original device under test, and where all three speed options are not available preset defaults can be supplied.

Determination of Distortion Characteristic at One or Multiple Attenuation Levels.

The final characteristic of the device to materially affect the audible performance is the non-linearity of the gain control element. Due to the analogue nature of many such gain elements significant non-linearities may be present both due to the device itself and due to the feedback of any audio signal into the gain control signal 16 of fig

1. The non-linearity of the device may be assessed in the following manner using and extending the techniques of The Prior Application.

There are three approaches to analysing and simulating the non-linearity of the gain element. One is to determine a single characteristic set of impulse responses according to the method of The Prior Application of the gain control element in the linear portion of the curve of the device under test. This may be performed by setting the threshold control of the device sufficiently high that the test sequence of impulses of the The Prior Application do not at any time cause any gain reduction to take place. This establishes a fixed analysis of the of the gain control device which may be used in the simulation as a fixed processing characteristic to be applied to an input signal in addition to the gain control adjustment already described.

A further method is to assess two characteristic sets of impulse responses of the gain control at two different attenuations. Since most gain control devices operate in the range of zero gain reduction to 20dB of gain reduction, it is desirable to take two measurements of the distortion of the gain control device at these two extremes and to linearly interpolate them during simulation using methods of bilinear interpolation as disclosed in The Prior Application and shown in figure 13, in which one dimension of interpolation is the instantaneous amplitude of the incoming signal being interpolated between adjacent impulse levels, and the second dimension is interpolation between non-linearities of the two gain characteristics according to the degree of attenuation to be applied.

A third method is to take multiple assessments of the distortion characteristic at multiple gain reductions and to pair-wise interpolate to get the desired characteristic for any given required gain reduction.

In order to take measurements of the gain control element at specific attenuations it is necessary to modify the test signal described in The Prior Application to ensure that each test impulse is applied at a known attenuation of the device under test.

In order to do this the device under test should be set to slow attack and decay characteristics and a large ratio, say 20:1 if available.

5 Using the techniques already discussed to assess the gain characteristic it is now possible to apply a sine wave tone, typically at 1kHz, that results in a known attenuation. This level is then maintained until the gain of the device under test has stabilised, again using the techniques already described.

10 Figure 7 (a) shows the sine wave being applied at 91 to achieve the desired gain. The sine wave is then removed (over short period T1 to minimise the impulsive effect). A period T2 is then allowed to elapse to allow any stored energy to dissipate. Figure 7(b) shows the output of the device under test which is being analysed by both the level detector 32 of figure 1 and the impulse storage unit 33. The signal 95 has been measured to indicate a steady state at the desired gain reduction. During period T2
15 any ripples 96 are ignored.

Step impulse 92 is applied after T2 and held as the response to this impulse 97 is stored. Finally after period T3 the step impulse is removed and the sine wave restored. The recovered wave 99 is then allowed to stabilise before the process is repeated for
20 the next different amplitude impulse.

In this way a complete set of impulses are sequentially applied to the device in a known stable gain state and stored and processed according to the method of The Prior Application. In this way a characterisation of the non-linear response of the gain
25 control element at the know gain reduction is obtained.

The sequence may be applied at a different desired gain to assess the non-linear response of the device at this different gain. In this way it is possible to store either two response sets for the two desired gains at either extreme of the range or if desired
30 a complete set of responses spanning the desired range which may be pair-wise interpolated during simulation.

In experiment it has been determined that gain control elements demonstrate comparatively short impulse responses and so the entire impulse response may be captured before the gain has significantly changed. In practice it is found a settling time T2 of about 100 samples at 48kHz is sufficient and a storage time T3 of 200 samples is sufficient to encapsulate the performance of the device under test. Thus the test is completed for each impulse in under 10mS from the discontinuation of the level setting tone 91. Recovery time constants of about 1 S are therefore sufficient to hold the gain sufficiently constant during this period.

It should be noted that if the make-up gain controls are not adjusted during taking a sequence of measurements and if the absolute levels of the impulse response are recorded, then the gain reduction as well as the non-linearity characteristic is encapsulated in the store impulse response. This fact is relevant to the discussion of simulation of the device.

Simulation of Level Control Device

Figure 9 shows an overall diagram of the simulator of level control devices.

Analogue to digital converter 101, which may be omitted if the input signal is already in digital form, takes an input analogue signal and feeds the signal to the modified convolution processor device 102 as described in The Prior Application comprising the modified convolution device 103 which convolves the incoming signal with the impulse response stored in the Finite Impulse Response Set (FIR set) storage device 105, under control of the amplitude assessment device 104 which selects on a sample by sample basis an appropriate response or pair of responses from FIR set storage 105 and also supplies an interpolation value 120 to processor 103.

The convolution processor 102 may also contain additional inputs 121 and 124 which receive a second interpolation value 121 (if required) and FIR set selector value 124 (if required), this FIR set selector value selecting between FIR sets representing

different gain reductions of the original device under test, and the interpolation value 121 providing interpolation values between them.

5 Address value 122 is used to select the appropriate offset in each FIR appropriate to the convolution operation and the amplitude assessment unit provides up to two FIRs from each set providing two values for interpolation, with set selection 124 selecting two sets of values to use, thus providing 4 data values on input 125 to the convolution processor for each step in the convolution. The two interpolation values 120 and 121 provide for bilinear interpolation between these four values selected on a sample by
10 sample basis of the input signal.

The input signal is further applied to the amplitude assessor with time constants 106 which determines an envelope for the input signal on a sample by sample basis under control of the user selected time constants 109. The resulting envelope 126 is fed to
15 gain reduction calculation device 127.

Adjustment is first made according to any user desired change in the threshold setting. If the original measurements were taken with an assumed threshold of -20dB, and the user now select a threshold of -10dB, the difference (10dB) is subtracted from the
20 input level derived in order to determine the gain variation required to simulate by reference to the derived gain characteristic data. In general if the original measurements of the gain characteristic were taken at a threshold of -A dB and the user now desires a threshold of -B dB the amount (A-B) dB is subtracted from the derived input signal envelope.

25

Gain table data according to figures 4 and 5 is stored in Gain Table Memory 128 and whenever the user selects a new ratio a table appropriate to the desired ratio is calculated according to the description above in reference to figure 6 and stored in memory 130.

30

The envelope of the input signal, modified by a user change in desired threshold, is fed to memory 130 to determine the desired output level for the signal of this envelope, and accordingly to select a desired gain for the signal at this instant.

- 5 Impulse set selector 107 selects the desired gain according to the method to be described which requires selection of an appropriate impulse response set or pair of them which is fed in signal 124 to the impulse response memory addressing, and (in the case where a pair of impulse responses or sets of them is used) a linear interpolation value which is fed in signal 121 to the convolution processor.

10

Where the gain required lies between that provided by two stored FIR sets the selector unit 107 selects the two FIR sets and determines the appropriate interpolation value 121. Where the gain lies outside the range of stored FIR sets the nearest is used and additional gain control is provided by additional gain calculator device 108.

15

User required make-up or overall gain may be applied by user selection 111 providing an extra gain demand signal which is combined in combiner 112 (which may be an adder if the gain signals are in logarithmic form, for example specified in dB, or a multiplier if specified in the form of a multiplicative value). The resultant gain signal 20 is fed to the additional gain element 113 which applies this gain to the output of the convolution device and feeds the resultant signal (by way of digital to analogue converter 114 if desired) to the output of the device.

Amplitude assessment with time constants

25

Amplitude assessment may be performed in a number of ways familiar to a skilled person. An example is shown in figure 10. The absolute value of an input sample S is determined and is known as |S|. This is compared with the current envelope E. If it is greater, the value E is increased according to the amount |S| exceeds E under 30 control of the attack time constant k1, as follows

$$E = E + k1 \cdot (|S| - E).$$

The release time constant is then applied which results in an exponential reduction in E:

5 $E = k_2 \cdot E.$

In general the attack time constant is much larger than the release time constant and so an approximation of the envelope of the input waveform is obtained with the desired time constants.

10

It is possible to elaborate this with two stages of time constant or with additional low pass filtering to remove unwanted audio modulation of the derived envelope with improvements in overall distortion characteristics but the above is sufficient for an acceptable simulation.

15

The time constants k_1 and k_2 may be related to analogue time constants T_1 and T_2 by noting that k_1 raised to the value r gives $\exp(-1/T_1)$ and k_2 raised to the value r gives $\exp(-1/T_2)$ where r is the sample rate in samples per second.

20 Gain Reduction Processor

By reference to the gain characteristic of the device under test an appropriate gain table is derived for the ratio desired by the operator according to the method already described. This may be calculated once each time the user selects a desired ratio
25 setting.

25

On a sample by sample basis the envelope signal derived from the amplitude assessment of the input signal is determined and converted to logarithmic form (i.e in dB relative to 0dBm input). If at this stage it is desired to adjust the threshold from
30 that used during the sample, an increase in desired threshold is subtracted from the envelope (a decrease is added) and the result used to determine a desired gain from

the table of output amplitudes versus input amplitudes. If the value falls between table entries a simple linear interpolation is applied.

Once the gain required is determined, it is now necessary to determine whether there is a FIR set stored for this gain value. If there is then this FIR set is selected into the convolution selection algorithm and no additional gain adjustment is generated. If there is not a FIR set for this gain reduction, but for two other gain reductions on either side of the required gain, these two FIR sets are selected into the convolution algorithm and an interpolation value generated to derive the desired gain.

If FIR set A gives an attenuation of a dB and FIR set B gives an attenuation of b dB, then the multiplicative factor appropriate to each M_a and M_b is as follows

$$M_a = 10^{(a/20)}, \text{ and } M_b = 10^{(b/20)}.$$

Applying an interpolation factor of j to FIR set A gives a total multiplicative gain M_g of

$$M_g = j.M_a + (1-j).M_b.$$

And the desired gain in dB $G = 20 \log(M_g)$, or $M_g = 10^{(G/20)}$.

Thus j may be derived for this sample from the rearrangement

$$j = (M_g - M_b) / (M_a - M_b). \quad (5)$$

M_a and M_b are known in advance for each FIR set so it is only necessary to calculate M_g and hence j for each sample. In addition a list may be kept at for example 0.1 dB intervals of all desired gains and the resulting choices of FIR sets and interpolating factors so that simple table look up may be performed on a sample by sample basis.

Finally the case where the desired gain is greater than or less than that for every stored FIR set. In this case the single FIR set closest to the desired FIR set is selected into the convolution algorithm and an additional gain factor F is generated such that

5 $F = G - A,$

where G is the desired gain at this instant and A is the gain of the nearest FIR set. The additional gain F in dB is added to any make up gain in 112 and passed to the additional gain element 113 to generate the desired overall gain.

10

EQ in the side chain

Where the system is used for sampling a complete audio processing channel with including a gain control element, there may be some equalisation applied between the
15 input and the level controlling device. This results in some input signals being dynamically adjusted differently from others depending on frequency spectrum. A similar effect is achieved if a level control provides EQ in the side-chain (i.e. between the main audio signal path and the level detection device of the level control device 1 of figure 1.

20

Both these variations can be handled by taking an additional analysis of the signal path between the input and the compressor device by attaching the analysis input of the system, for example the analogue to digital converter of analyser 3 in figure 1, and taking a single impulse response test of this path to determine the frequency
25 characteristic. This characteristic may be replicated in the optional additional EQ unit 129 of figure 9.

This additional equalisation is not usually critical to the operation of the overall system and it may be implemented by a straight finite impulse response convolution
30 with the data derived from the above test or may be simulated by a derived infinite impulse response equaliser of broadly similar frequency response.

The existence of equalisation prior to the gain control element or in the side chain may suggest alternative frequencies for use in the gain reduction analysis such that the test signal is passed properly into the gain determining elements. It is also possible to use a number of different test frequencies to analyse the behaviour of the signal path prior to the compressor without reconnecting the test arrangement of figure 1 for subsequent synthesis.

Use without the Non-Linear effects synthesis of The Prior Application

Although this system has been described using the non-linear effects analysis and synthesis of The Prior Application it is possible to determine a single impulse response at each gain setting measured to determine frequency response and relative gain and apply the above technique without the non-linear section of the algorithm. In this way a restricted but useful synthesis of the device is achieved simulating the gain characteristic, the frequency response and variation with gain reduction (if multiple gain reductions are sampled) and also simulations of time constant if implemented. This may even be desirable if it is determined that the non-linearity of the original device is not desirable in a given simulation so in any event the degree of non-linearity should be made variable as described in The Prior Application and also to include using a single impulse response from a derived set of data.

It is also possible to omit the impulse response determination completely and to use the methods of the invention to simulate the gain characteristic alone or in combination with the time constant characteristic.

Use without Analyser or with data derived from alternate uses of the analyser

Although the description explains how to take measurements of existing audio dynamics processor devices and how to simulate them, it is also possible to generate the necessary data for synthesis by means of any desired model of such a processor or by mixing data obtained from different sources.

In particular:

1. The curves of gain characteristic can be calculated for a computer model of a desired effects device, or may be drawn by an operator on a computer screen either completely freely, or may be modified manually by graphical manipulation from existing data. In any of these ways an operator may obtain a desired gain characteristic not available in a real dynamics processor device.
2. Time constant characteristics may be freely substituted for any measured characteristics, or derived by operator choice to more closely suit the audio material to be processed.
3. Characteristic FIR sets (either linear or non-linear) may be generated by computer model of a gain control device that it is required to simulate, for example there are now a number of ways to simulate valve circuits, and it would be possible to generate arithmetically the impulse response of these simulations appropriate to a variety of levels of impulse and gain variations of the simulated device. These impulse responses may then be used with the methods described herein to process audio according to these derived characteristics.
4. Characteristic FIR sets (either linear or non-linear) may also be derived from other audio devices or systems which have a desirable characteristic and used with the methods described herein to simulate a dynamic control device not currently achievable.

One example is illustrated by reference to figure 11 in which it is shown how to derive two or more sets of impulse response characteristics from a microphone positioned in front of a loudspeaker 142 in a room 143 of a particular ambience. Responses (either linear using a single impulse test or non-linear as described in The Prior Application) can be obtained with the microphone at location 140 and at a different location 141 at a different distance from the loudspeaker, giving an amplitude reduction characteristic derived from moving away from the sound source.

The two impulse responses or sets of them embody two different gains (as well as different ambience characteristics) and this pair of impulse response characteristics is now used in the method of simulation described herein to simulate compression and the compression effect simulated will be one of reducing dynamics by moving
5 towards and away from the audio source according to amplitude of the source. This may be used for example to simulate the effect of a singer who has learned to moved to and from a microphone according to the volume of the voice to control dynamics.

A further example is illustrated in figure 12 in which a set of non linear impulse
10 responses of a system 150 (which may also be stereo or other multi-channel system) is taken at progressively increasing levels comprising at least two different sets of levels 151 and 152 according to The Prior Application to generate response sets 153 and 154 where the system 150 exhibits gain reduction through progressively moving into non-linearity. Each FIR set is assigned a gain value for use in the gain reduction
15 simulation according to an estimate of the gain of the system at the peak signal level, typically by estimating the peak deviation of the impulse response obtained from each sequence of test impulses. These are used as a set of non linear FIR sets of at least two intrinsic gains in the simulation method described herein to produce a compression effect characteristic of an overdriven system. System 150 may be a
20 totally electronic device or for example an amplifier and speaker system with sound picked up by microphone.

By reference to figure 13 the exact operation of the bi-linear interpolation of figure 9 can be demonstrated.

25 FIR set storage 105 of figure 9 is also illustrated in figure 13 as a sequence of storage of sets of impulse responses. In this case each impulse response is shown with 8 elements for clarity but in practice many more elements will be used.

30 According to The Prior Application a factor k is determined to interpolate between two impulse responses in a FIR set (see for example figure 10 of The Prior Application). This is shown appearing in figure 13 where it is used to interpolate

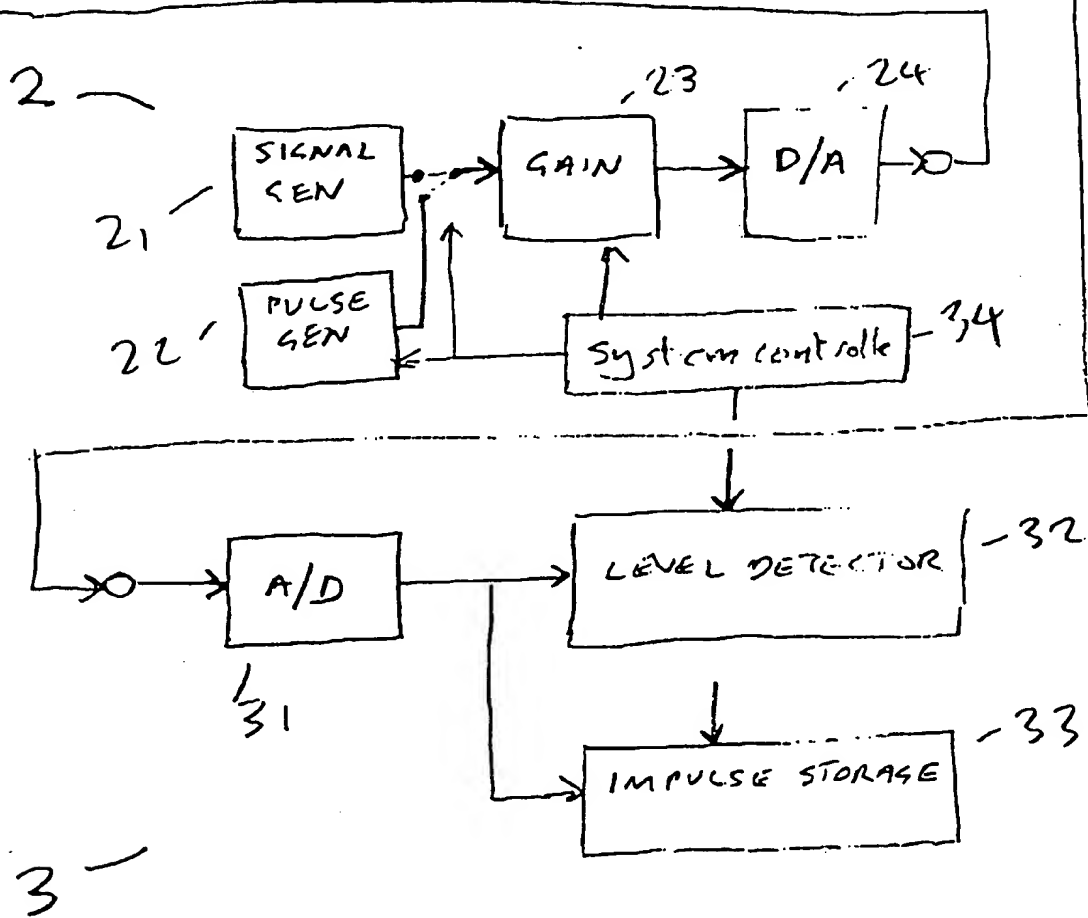
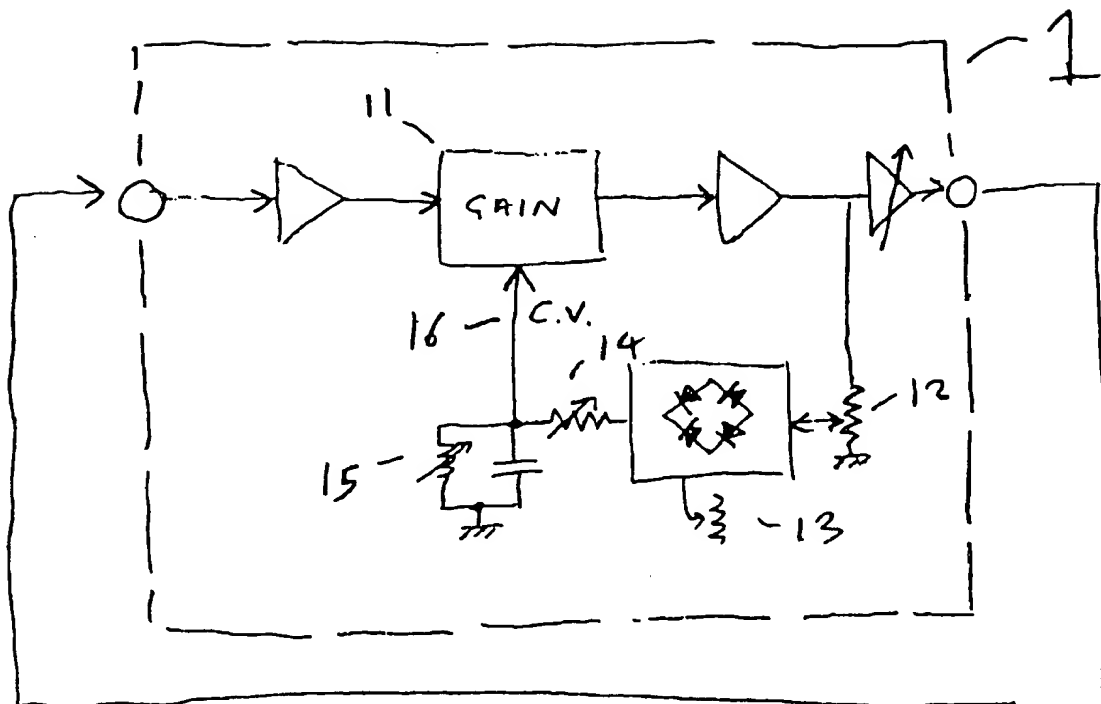
between elements of for example response #4 and #5 of FIRSET 1, and also between the matching elements of FIRSET 2. It is also shown in figure 9 at 120.

5 Interpolation factor j is also determined in equation (5) above as the interpolation between two sets of impulse responses representing two different gains. These are illustrated in figure 13 as FIRSET 1 and FIRSET 2.

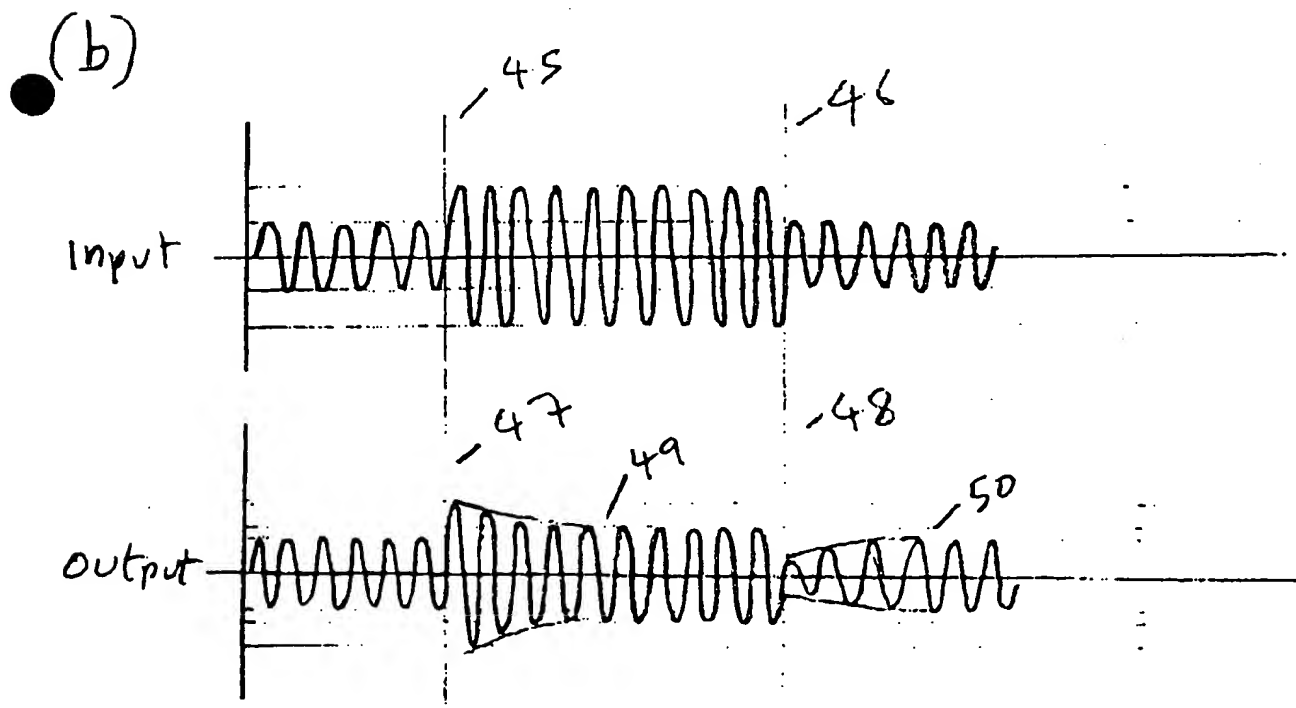
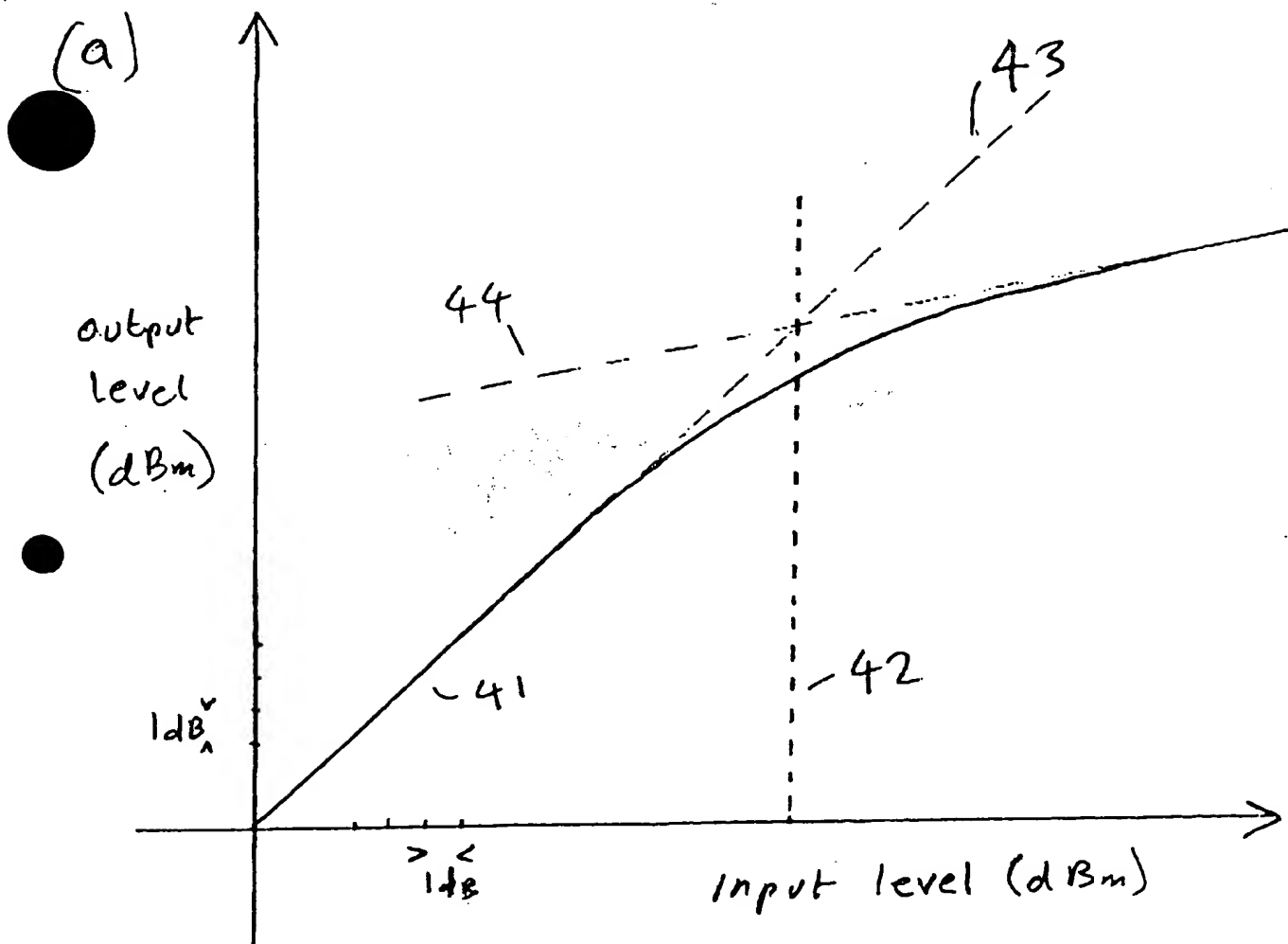
10 In this way a single value 160 is derived from the 4 elements of the 4 impulse responses by bilinear interpolation using values j and k which are determined on a sample by sample basis for each step in the convolution process as the elements of each impulse response are stepped through in the convolution algorithm.

15 In practice, as described and shown in figure 14 of The Prior Application, it is convenient to generate part sums for each multiply accumulate operation for a given input sample, where j , k and the pointers into the FIR set memory can be determined once for each sample and applied repeatedly. In this way for each input sample an output sample can be produced and part sums for subsequent output samples assembled.

20 -end-



This Page Blank (uspto)



2/13

Fig 2

This Page Blank (uspto)

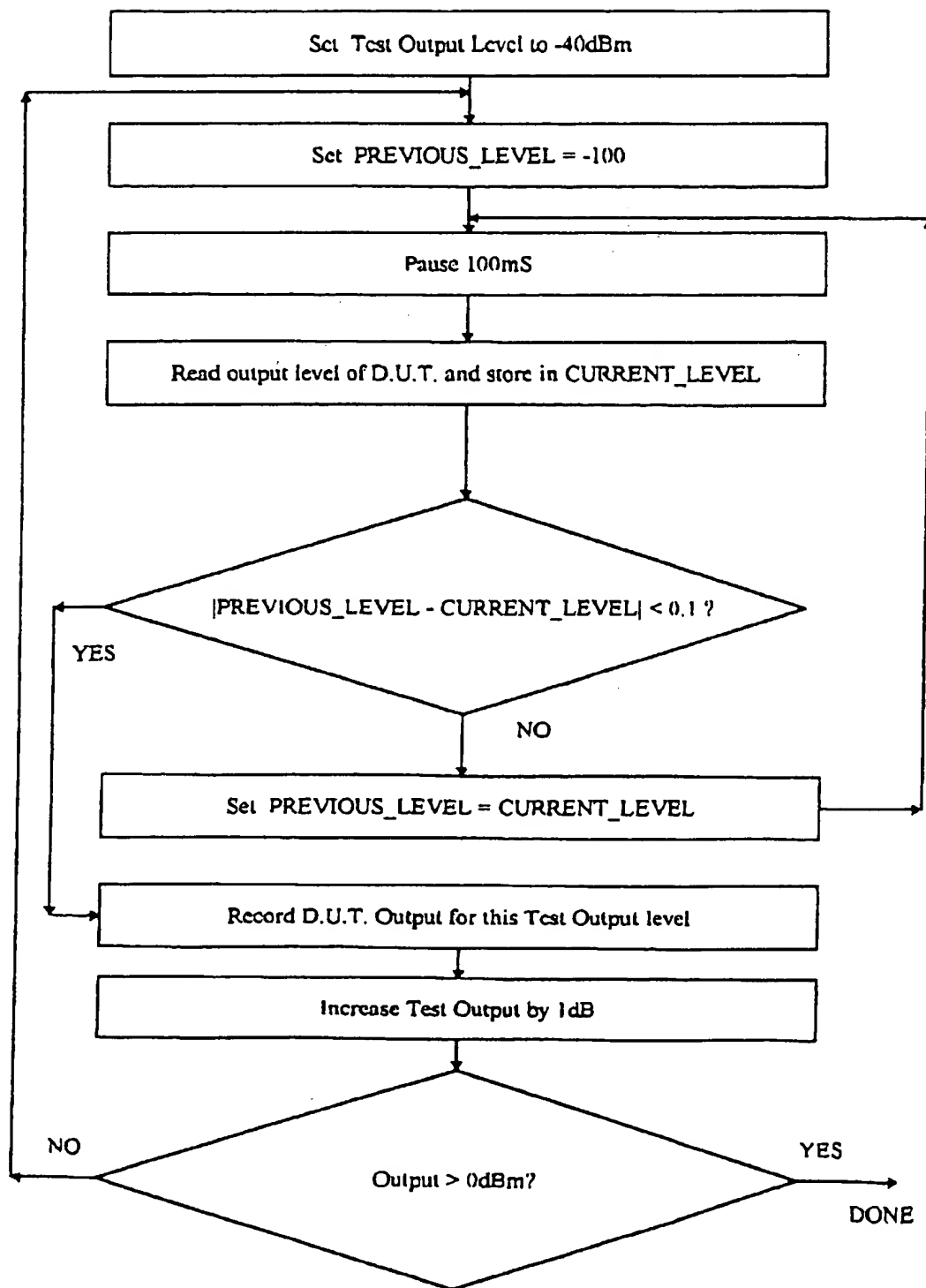


Figure 3

This Page Blank (uspto)

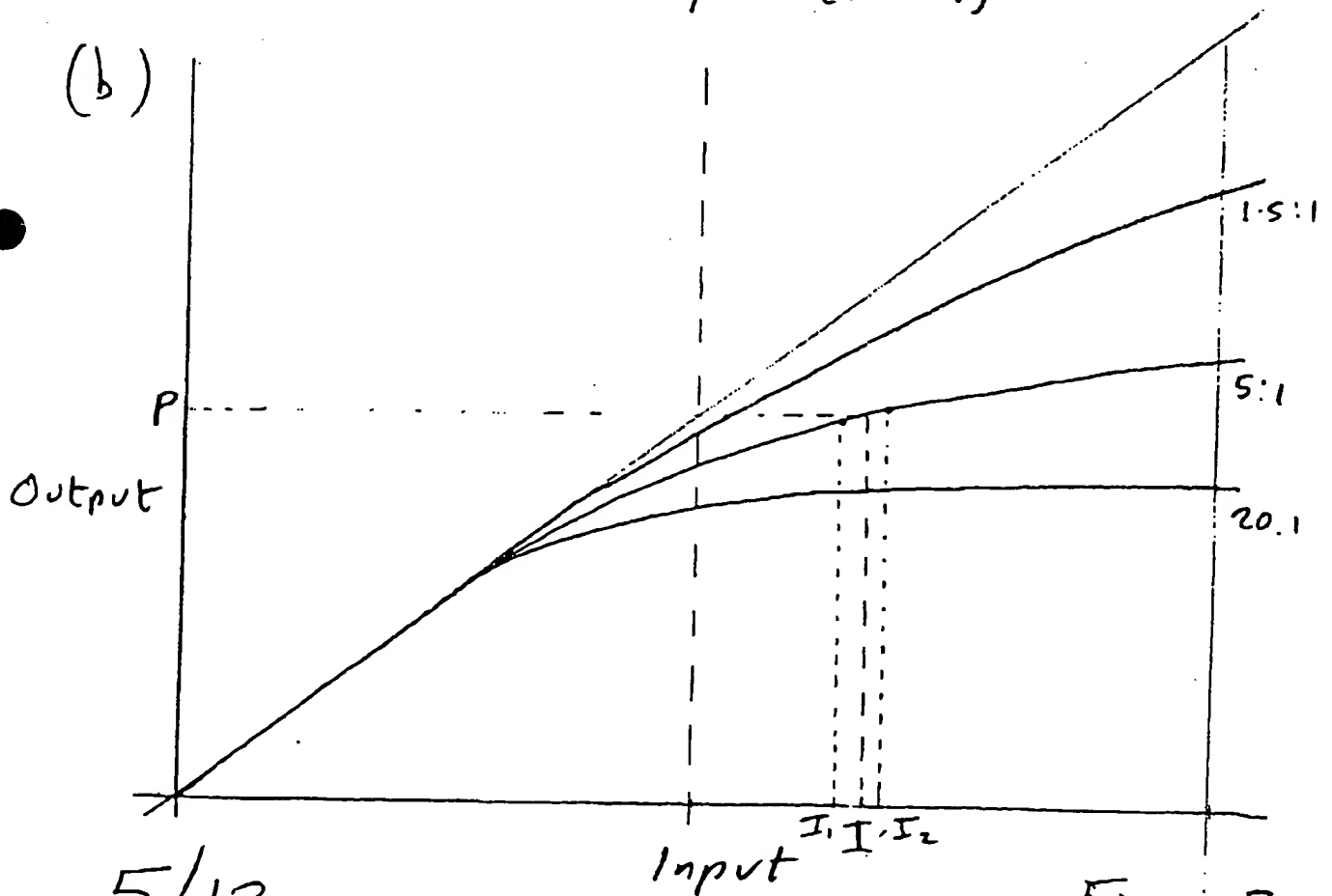
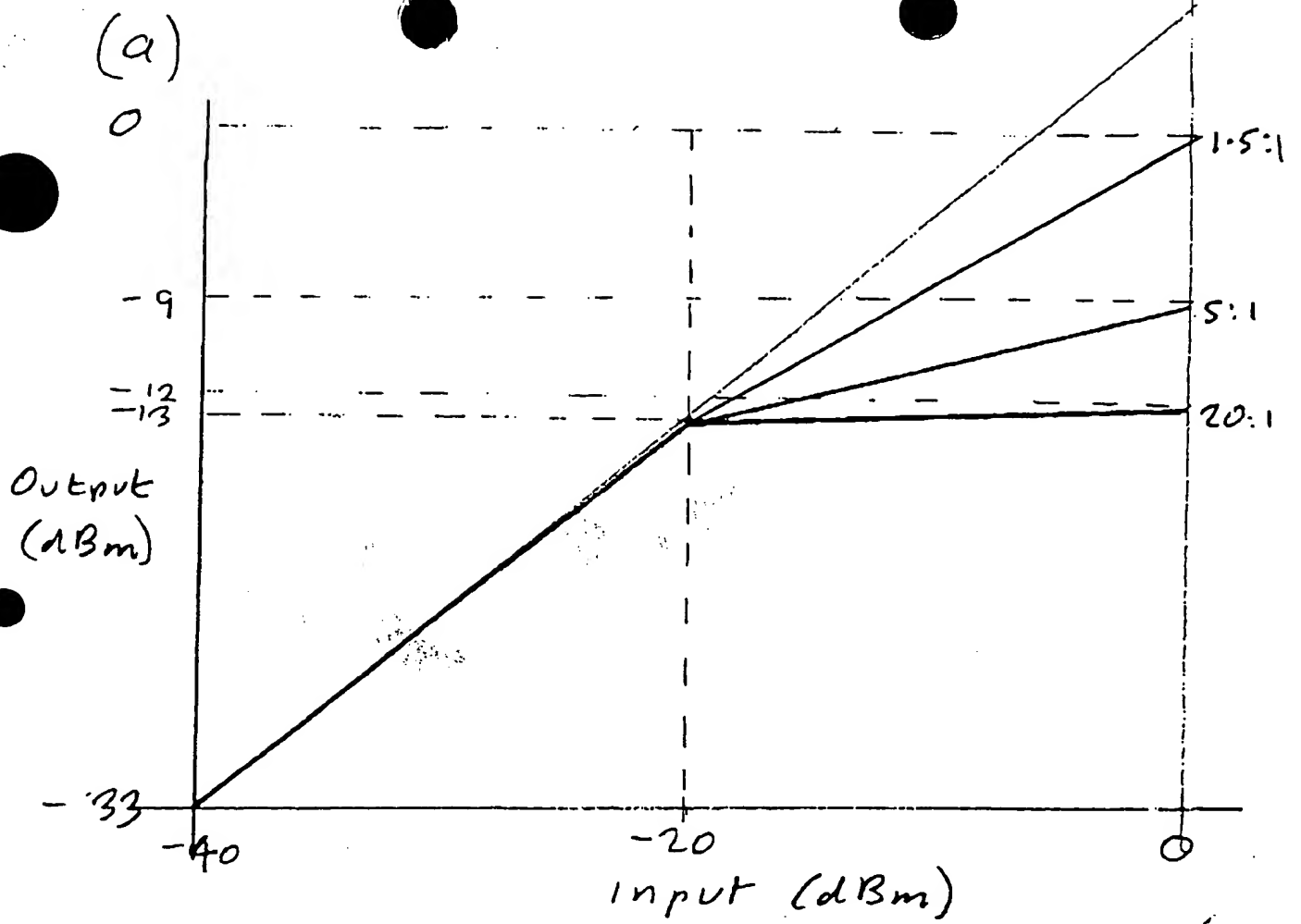
Figure 4 (a): Table of Output Level Test Data for Transfer Curve of 'Ideal' Compressor

input / ratio	1.5:1	5:1	20:1
-40dBm	-33.3	-33.3	-33.3
-39dBm	-32.3	-32.3	-32.3
-38dBm	-31.3	-31.3	-31.3
...			
-30dBm	-23.3	-23.3	-23.3
...			
-20dBm	-13.3	-13.3	-13.3
...			
-10dBm	-6.7	-11.3	-12.8
...			
-2dBm	-1.3	-9.7	-11.9
-1dBm	-0.7	-9.5	-11.95
0 dBm	0.0	-9.3	-11.8

Figure 4 (b): Table of Output Level Test Data for Transfer Curve of 'Typical' Compressor

input / ratio	1.5:1	5:1	20:1
-40dBm	-35.5	-35.7	-35.5
-39dBm	-34.5	-34.6	-34.5
-38dBm	-33.4	-33.5	-33.4
...			
-30dBm	-24.9	-25.5	-27.2
...			
-20dBm	-16.3	-17.5	-19.2
...			
-10dBm	-11.2	-13.4	-15.8
...			
-2dBm	-1.7	-11.7	-12.8
-1dBm	-1.1	-11.6	-12.8
0 dBm	-0.3	-11.5	-12.7

This Page Blank (uspto)



5/13

Figure 5

This Page Blank (uspto)

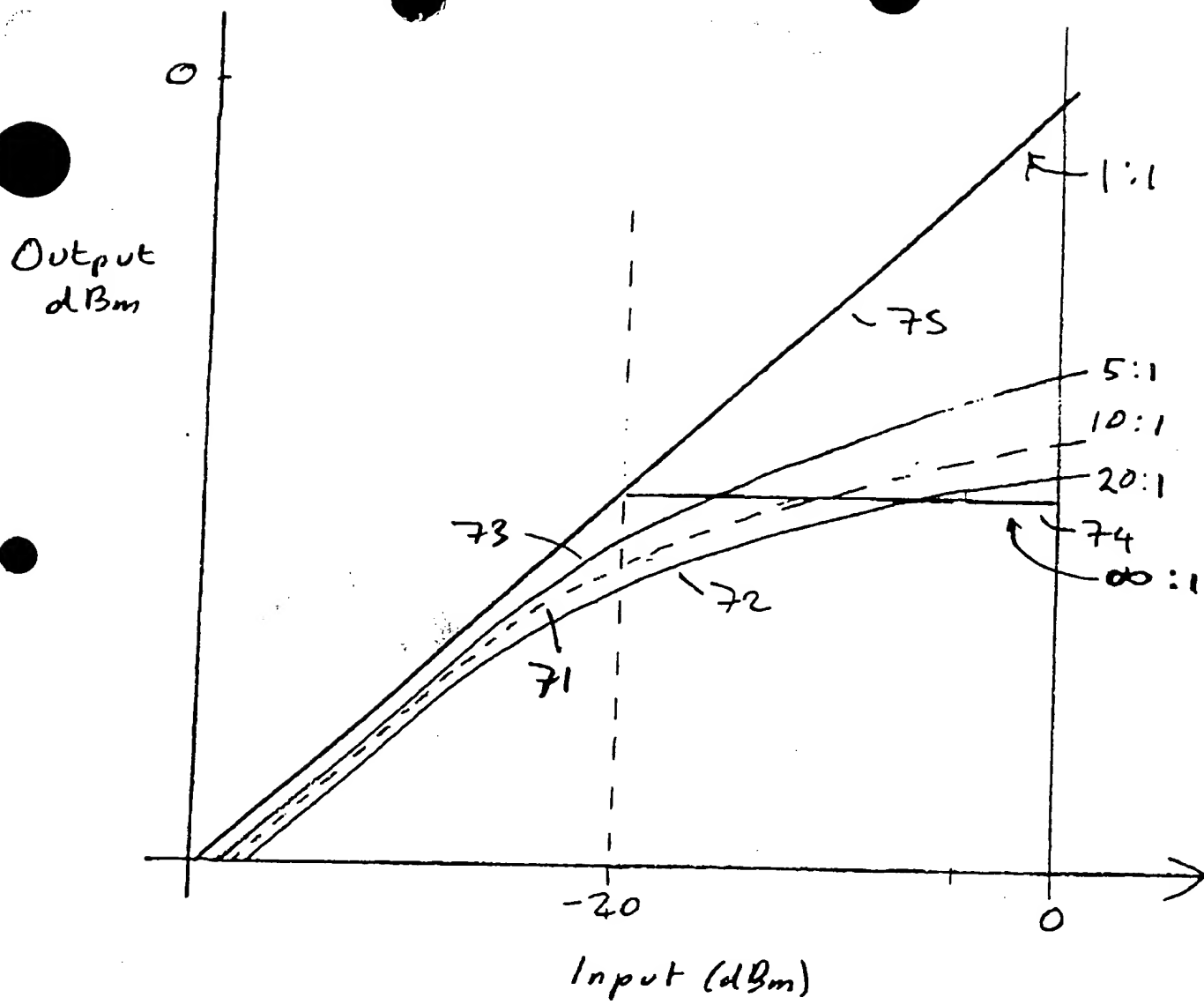
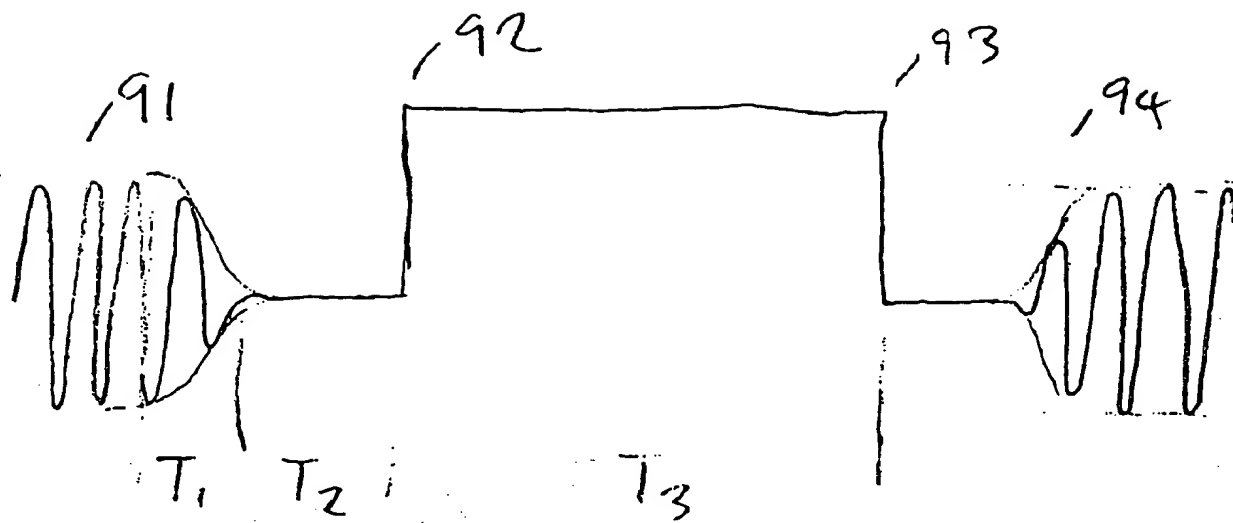


Figure 6

This Page Blank (uspto)

(a)



(b)

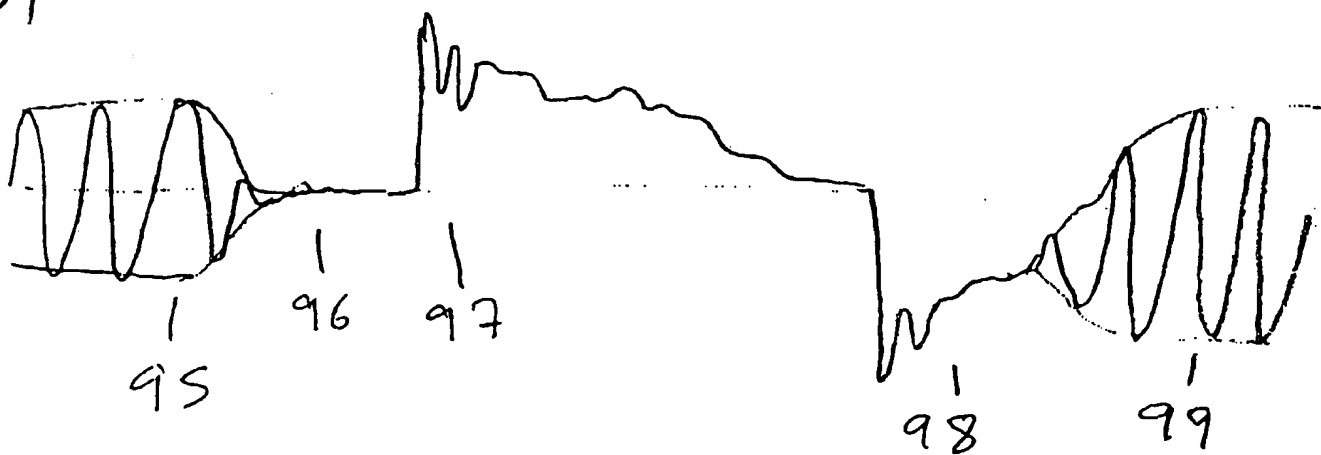


fig 7

This Page Blank (uspto)

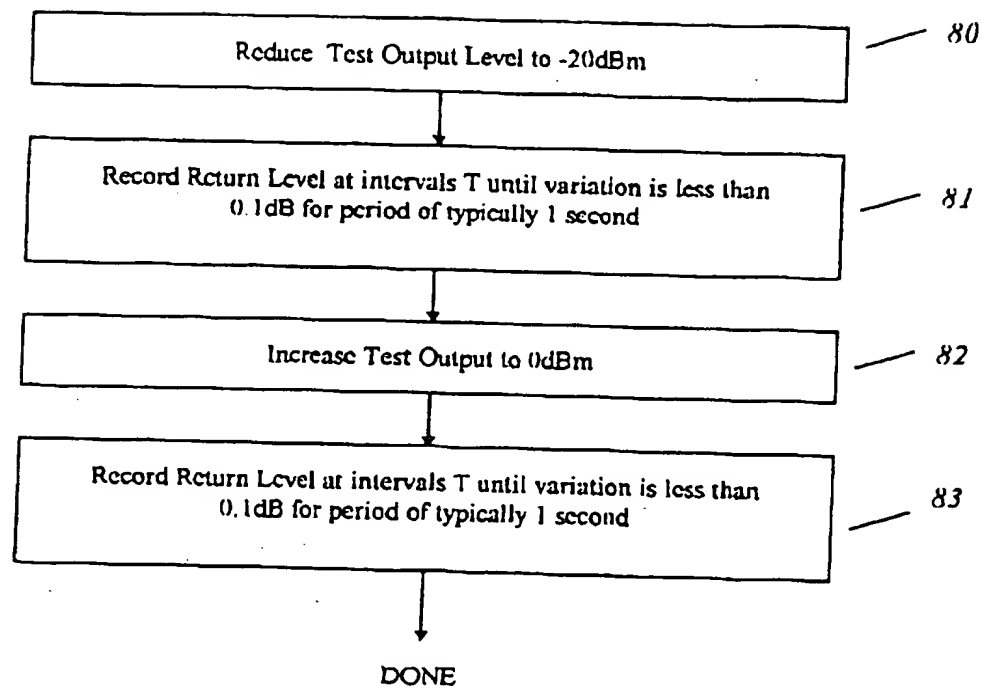


Figure 8

This Page Blank (uspto)

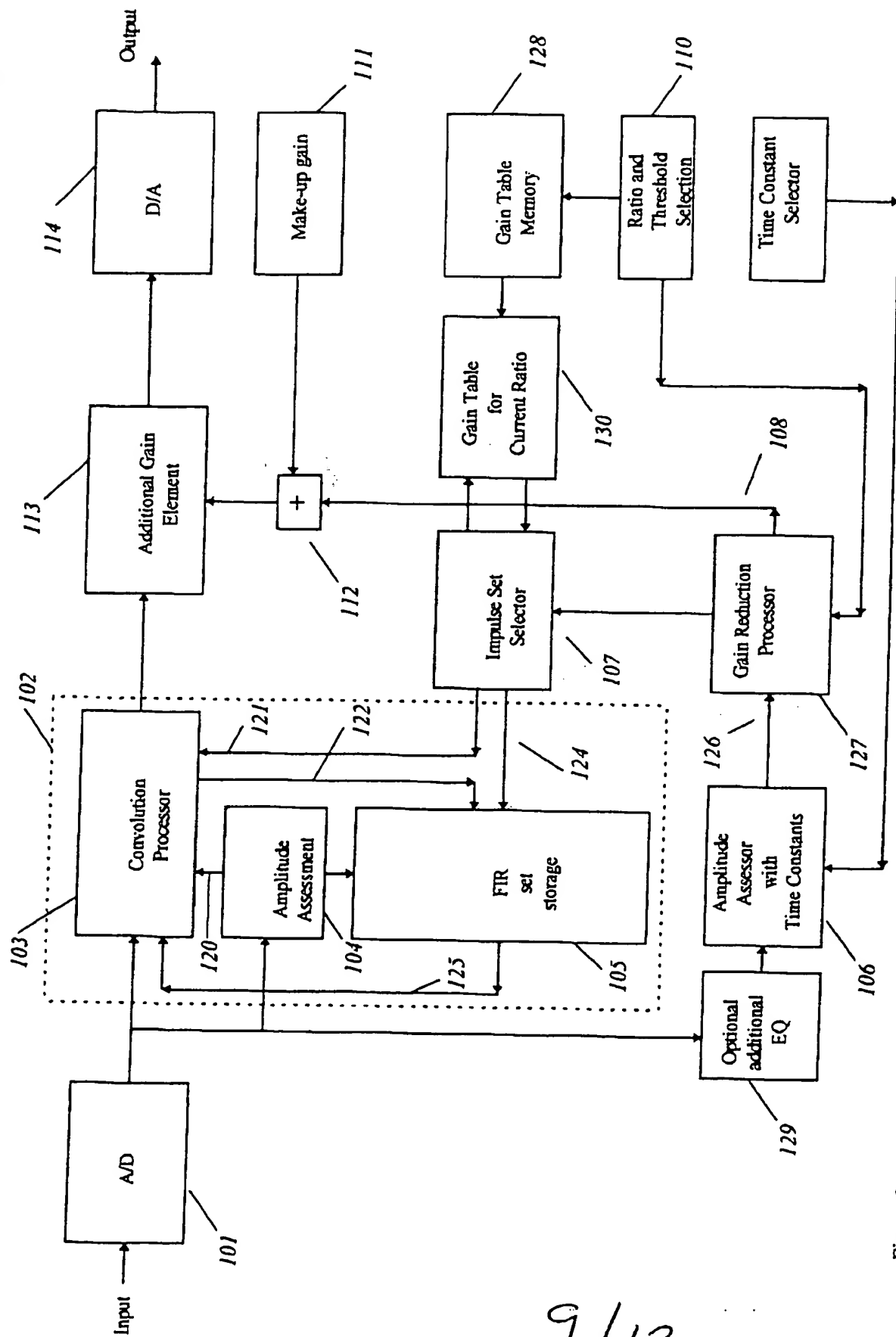


Figure 9

This Page Blank (uspto)

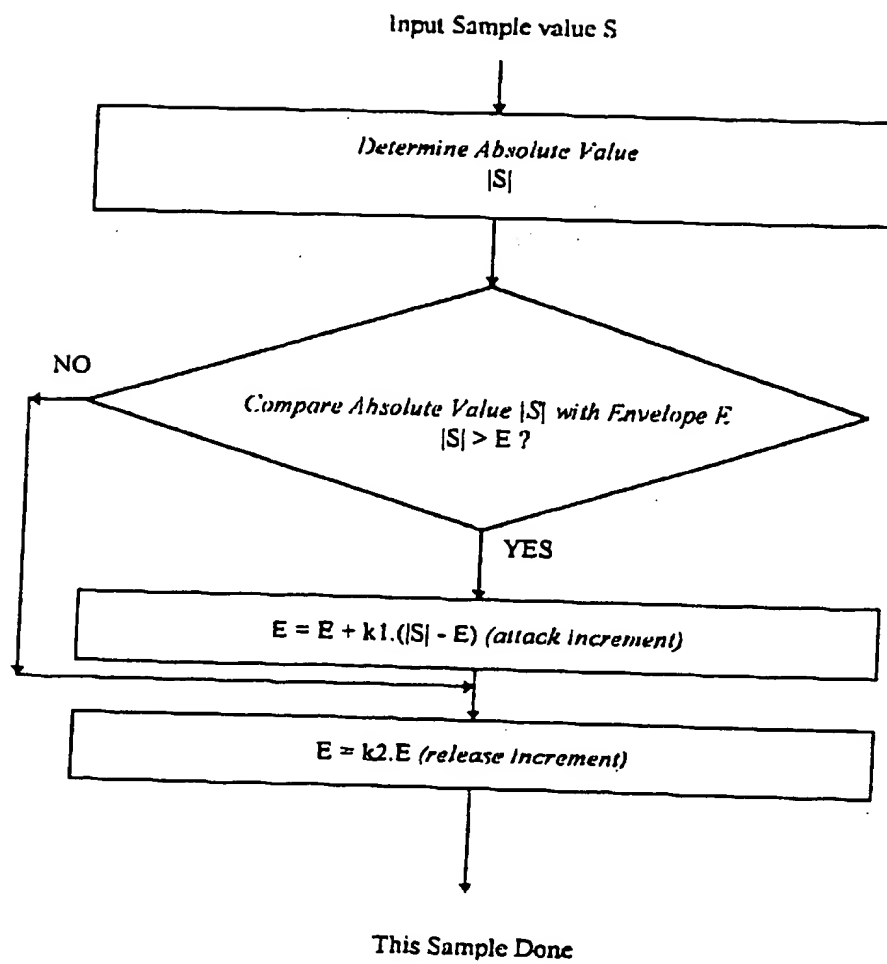


Figure 10

10/13

This Page Blank (uspto)

11/13

-143

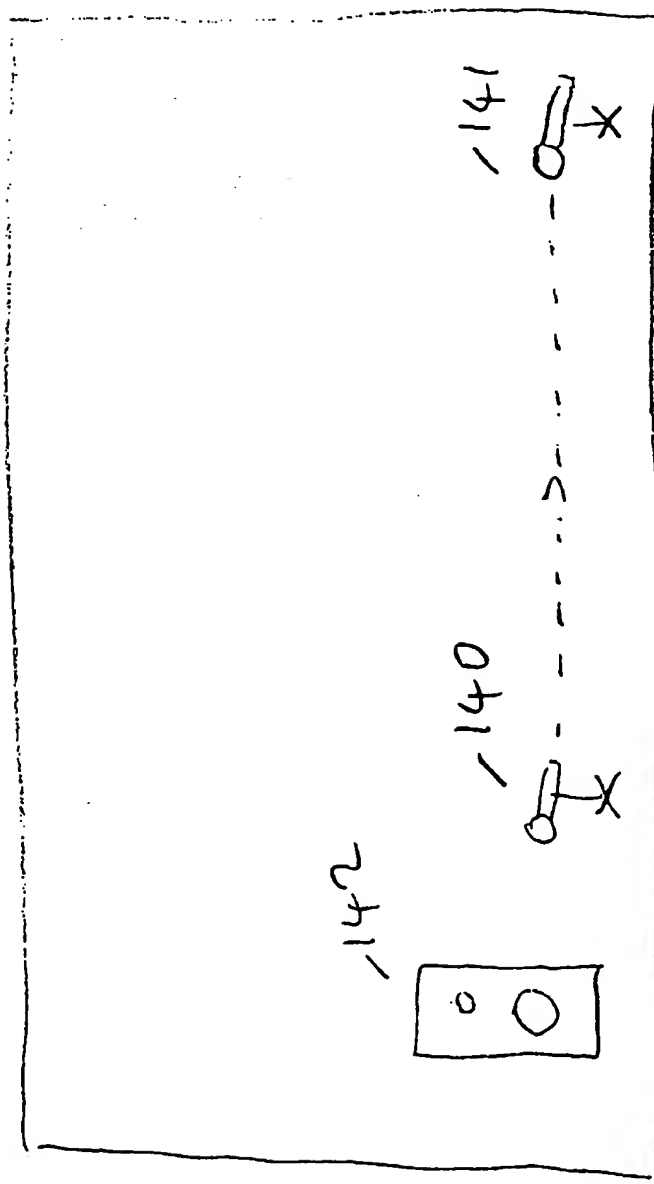


Figure 11

This Page Blank (uspto)

12/13

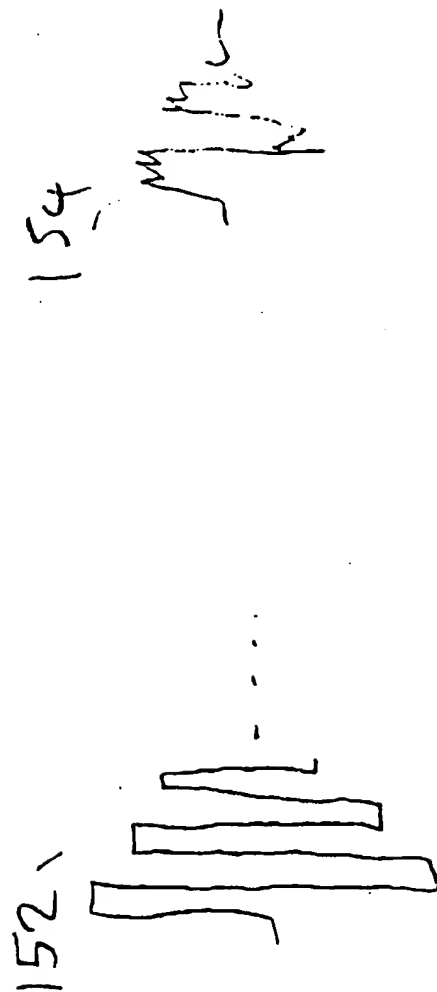
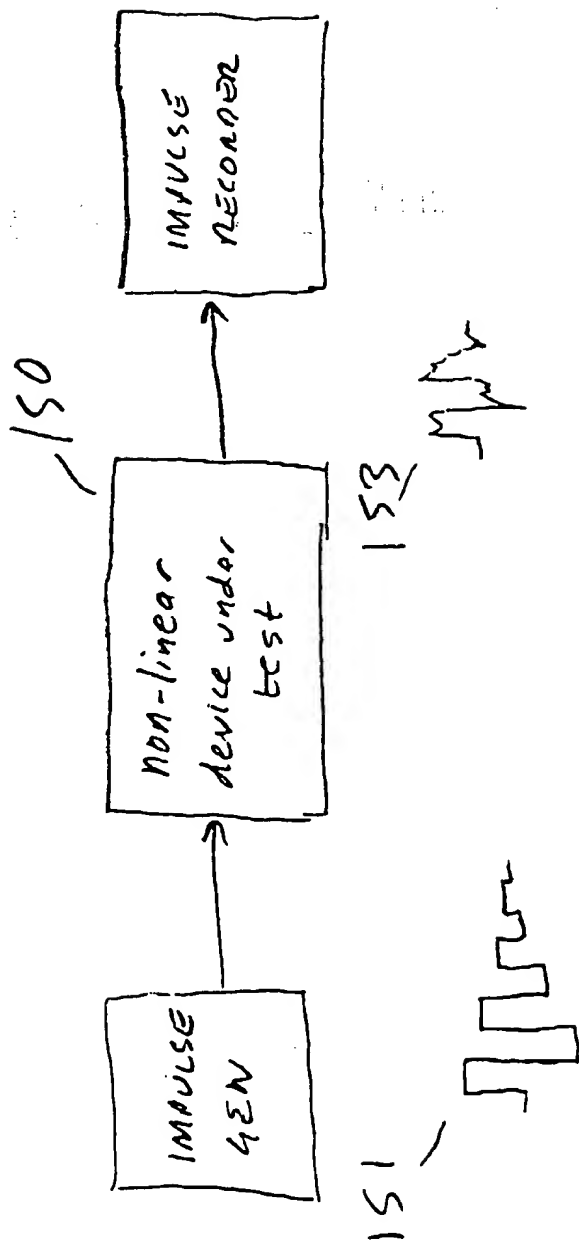


Figure 12

This Page Blank (uspto)

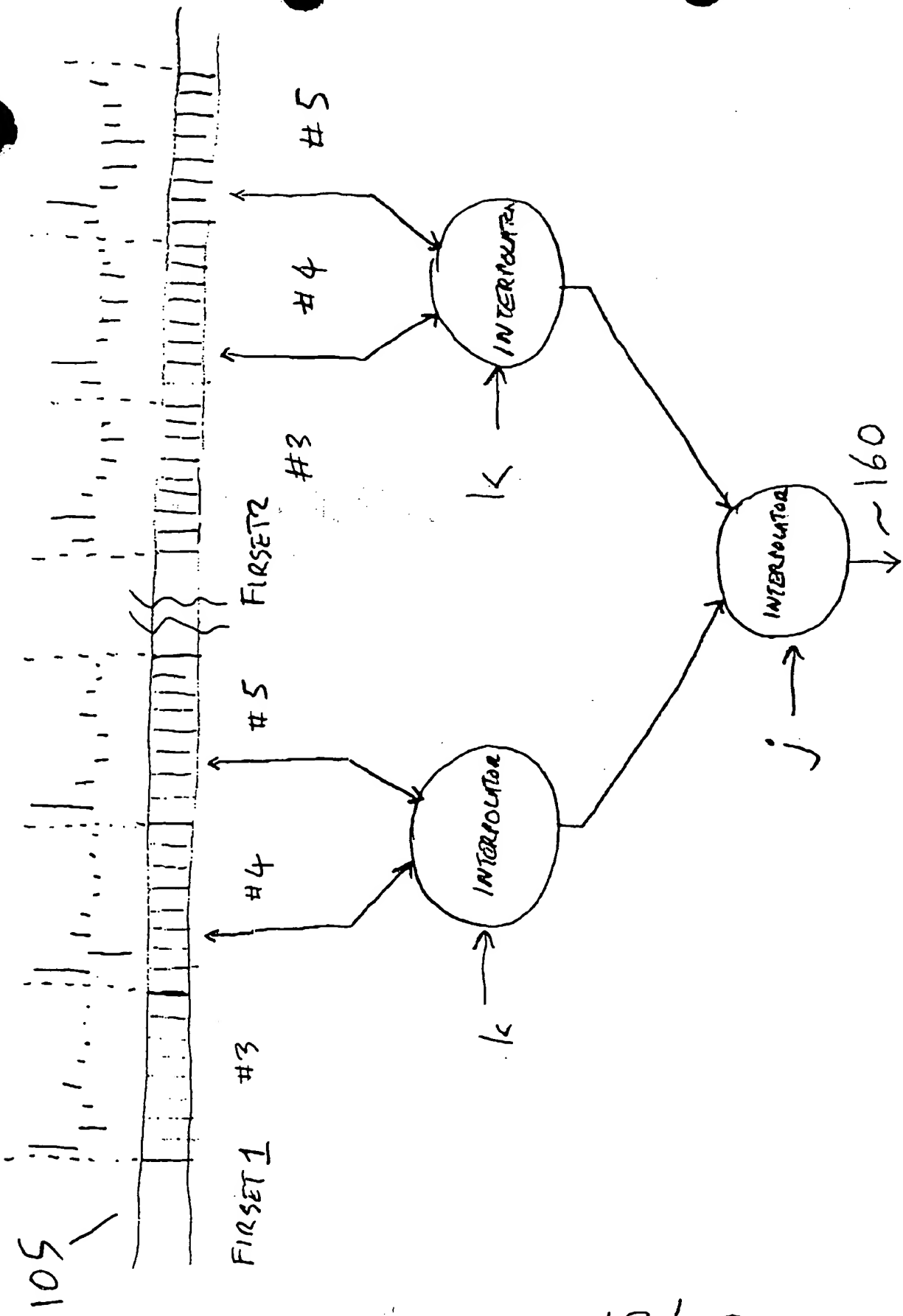


Figure 13

PCT / GB99 / 03753

6.1.00

Reddie & Grose.

This Page Blank (uspto)